REMARKS

I. Introduction

In response to the Office Action dated January 20, 2010, claims 1, 2, 10, 11, 19 and 20 have been amended, claims 3, 9, 12, 18, 21 and 27 previously canceled, and new claims 28-33 have been added. Claims 1, 2, 4-8, 10, 11, 13-17, 19, 20, 22-26 and 28-33 are now pending in the application. Re-examination and re-consideration of the application, as amended, is requested.

II. Claim Amendments

Applicants' attorney has made amendments to the claims as indicated above. Unless otherwise indicated, these amendments were made solely for the purpose of clarifying the language of the claims, and were not required for purposes of patentability or to distinguish the claims over the prior art.

III. Examiner Interview

Reference is hereby made to a telephone interview between Applicants' attorney Victor G. Cooper, and Examiner Faulk in connection with the present patent application on January 15, 2010. Potential claim amendments were discussed that might bring the case to allowance, but no agreement was reached. The McDowell reference was also discussed and compared to the Applicant's claims.

IV. The Cited References and the Subject Invention

A. The McDowell Reference

U.S. Patent No. 6,931,370, issued August 16, 2005 to McDowell discloses a system and method for providing interactive audio. The disclosed system and method is said to provide low cost fully interactive immersive digital surround sound environment suitable for 3D gaming and other high fidelity audio applications, which can be configured to maintain compatibility with the existing infrastructure of Digital Surround Sound decoders. The component audio is stored and mixed in a compressed and simplified format that reduces memory requirements and processor utilization and increases the number of components that can be mixed without degrading audio quality. Techniques are also provided for "looping" compressed audio, which is an important and standard feature in gaming applications that manipulate PCM audio. In addition, decoder sync is ensured by

transmitting frames of "silence" whenever mixed audio is not present either due to processing latency or the gaming application.

B. The Friedman Reference

U.S. Patent No. 5,337,041, issued August 9, 1994 to Friedman discloses a personal safety guard system enabling a guardian or caretaker of a person or pet to transmit an alarm condition signal from a hand-held unit carried by the guardian. When the alarm condition signal is received by a portable alarm unit adapted to be worn by the person or pet under the guardian's supervision, the alarm unit operates to alert the wearer that its guardian is looking for them, and to alert others nearby that the wearer is in need of assistance by producing a number of different alarm indicators. The alarm indicators produced by the portable alarm unit include an intelligible voice message such as "Help, I'm lost" which is alternately sounded with a loud alarm sound, and flashing strobe lights. These alarm indicators, together with a confirmation signal transmitted from the alarm unit to the guardian's unit, enable the guardian to track and find their charge.

C. The Petrillo Reference

U.S. Patent No. 6,429,779, issued August 9, 1994 to Petrillo et al. discloses a telephone line monitoring and alarm apparatus. The apparatus is capable of continuously monitoring telephone line status and activating an audio-visual alarm if the telephone line becomes inoperative, incorporates an integral telephone plug to permit direct plug-in connection of the apparatus to a standard telephone wall jack as a self-contained and autonomous unit without the use of interconnecting cables or cable-plug attachments. An integral dual telephone jack splitter permits the uninterrupted use of standard telecommunication equipment, such as telephones, answering machines, or facsimile equipment while the apparatus monitors telephone line integrity. Micropower circuitry derives electrical power from the telephone line to provide visual ON status indication, and from a battery source independently of the telephone communication line voltage, to provide continuous telephone line monitoring, audio-visual alarms and audio-visual low battery voltage indication. The invention is intended to provide telephone subscribers with effective early detection and warning if the telephone line becomes inoperative due to intentional or accidental disruption of telephone service.

D. The Fiocca Reference

U.S. Patent No. 5,625,743, issued April 29, 1997 to Fiocca discloses a system for calculating a signal-to-mask ratio (806) for a subband in a subband in a subband audio encoder is calculating a signal level for each of the subbands based on an audio frame (604). Then, the masking level is calculated for the particular subband based on the signal levels, an offset function, and a weighting function (606).

E. The Kallergis Reference

U.S. Patent No. 4,934,483, issued June 19, 1990 to Kallergis discloses a method of reducing the overflying noise of airplanes having a propeller driven by a piston engine. The propeller is arranged on the engine shaft in such a way that positive components of the engine sound pressure fall on negative components of the propeller sound pressure. It is preferable to use an engine/propeller combination in which the number of engine ignitions per revolution of the propeller shaft divided by the number of the propeller blades is an integer, preferably being equal to 1.

F. The Smith Reference

U.S. Publication No. 2002/0173864 published November 21, 2002 by Smith discloses an automatic volume control for voice over Internet. The method includes: estimating an average frame volume estimate (VE) for each frame of data; calculating from a plurality of successive frame volume estimates at least one moving average of the volume estimates; comparing at least one of the moving averages with a known desired level that is associated with a psychoacoustically desirable audio volume level; calculating, independently of any compression applied to the data frame during encoding, a digital gain factor based upon the results of the aforementioned comparison; and adjusting a volume level of the audio data based upon the digital gain factor.

G. The Pai Reference

U.S. Patent No. 6,801,886, issued October 5, 2004 to Pai et al. discloses a system for improved digital data compression in an audio encoder. A threshold is established which depends on the bit rate of the input data. A determination is made whether the bit rate is above or below the established threshold. A masking index is calculated for the input data according to a first formula if the input data is being transmitted at a rate at or below the threshold. A second formula is used to calculate the masking index if the input data is being transmitted at a rate above the threshold. The masking index is used to generate a masking threshold, and data deemed insignificant relative to the masking threshold is ignored. A psychoacoustic modeler, which is included in the encoding section of an encoding/decoding (CODEC) circuit, is used to determine a masking index. The masking index is then used to generate a masking threshold. A masking threshold is an information curve generated for and unique to each piece of audio data which enters the CODEC circuit. The psychoacoustic modeler uses experimentally determined information about human hearing and, through a process called perceptive encoding, determines which parts of the input audio data will not be perceived by the human ear. The masking threshold is a curve below which the human ear cannot perceive sounds. The psychoacoustic modeler compares the masking threshold uniquely generated for the specific piece of input audio data and compares the masking threshold to the input audio data. This comparison dictates to the encoding section of the CODEC circuit which of the tones and noises contained within the input audio data can be ignored without sacrificing sound quality.

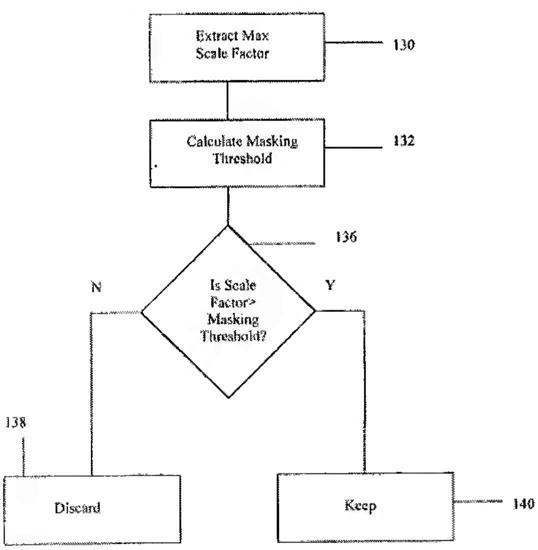
V. Office Action Prior Art Rejections

In paragraph 4, the Office Action rejected claims 1, 8, 10, 17, 19 and 26 under 35 U.S.C. § 103(a) as unpatentable over McDowell in view of Friedman and in further view of Petrillo. Applicants respectfully traverse these rejections.

With Respect to Claim 1: Claim 1 recites:

> *A method of automatic measurement of audio presence and level by direct processing of a data stream representing an audio signal in a processor, comprising:*
>
> *(a) extracting, in the processor, sub-band data from the data stream;*
>
> *(b) dequantizing and denormalizing, in the processor, the extracted sub-band data;*
>
> *(c) measuring, in the processor, an audio level for the dequantized and denormalized sub-band data without reconstructing the audio signal using channel characteristics;*
>
> *(d) comparing, in the processor, the measured audio level against one or more thresholds; and*
>
> *(e) triggering, in the processor, an alarm as determined by the comparing step (d), wherein the thresholds are set to generate the alarm based on: (1) loss of the audio signal in the data stream or (2) when an average level of the audio signal in the data stream is too high or too low, in order to monitor the audio presence and level within the data stream and to adjust the audio level as desired.*

Claim 1 recites the step of dequantizing and denormalizing the extracted sub-band data. According to the Office Action, McDowell teaches this step as follows:



Second, DTS Interactive unpacks the scale factors (step 120) and uses them in a simplified psychoacoustic analysis (see FIG. 9) to determine which of the audio components selected by the map function (step 54) are audible in each subband (step 124). A standard psychoacoustic analysis that takes into account neighboring subbands could be implemented to achieve marginally better performance but would sacrifice speed. Thereafter, the audio renderer unpacks and decompresses only those subbands that are audible (step 126). The renderer mixes the subband data for each subband in the subband domain (step 128), recompresses it and formats it for packing as shown in FIG. 4 (item 86).

However, the McDowell reference discloses a system wherein the bit allocation data used to define the quantization is fixed (in the interest of computational efficiency).

> More specifically, the components are preferably encoded into a subband representation, compressed and packed into a data frame in which only the scale factors and subband data change from frame-to-frame. This compressed format requires significantly less memory than standard PCM audio but more than that required by variable length code storage such as used in Dolby AC-3 or MPEG. More significantly this approach greatly simplifies the unpack/pack, mix and decompress/compress operations thereby reducing processor utilization. In addition, fixed length codes (FLCs) aid the random access navigation through an encoded bitstream. High levels of throughput can be achieved by using a single predefined bit allocation table to encode the source audio and the mixed output channels. In the currently preferred embodiment, the audio renderer is hardcoded for a fixed header and bit allocation table so that the audio renderer only need process the scale factors and subband data.

This limits the sub-bands to fixed (rather than adaptive) intervals, but this feature makes sense in McDowell because they are not considering psychoacoustic effects from sub-band to sub band.

> Second, DTS Interactive unpacks the scale factors (step 120) and uses them in a simplified psychoacoustic analysis (see FIG. 9) to determine which of the audio components selected by the map function (step 54) are audible in each subband (step 124). A standard psychoacoustic analysis that takes into account neighboring subbands could be implemented to achieve marginally better performance but would sacrifice speed. Thereafter, the audio renderer unpacks and decompresses only those subbands that are audible (step 126). The renderer mixes the subband data for each subband in the subband domain (step 128), recompresses it and formats it for packing as shown in FIG. 4 (item 86).
>
> The computational benefits of this process are realized from having to unpack, decompress, mix, recompress and pack only those subbands that are audible. Similarly, because the mixing process automatically discards all of the inaudible data, the gaming programmer is provided greater flexibility to create richer sound environments with a larger number of audio components without raising the quantization noise floor. These are very significant advantages in a real-time interactive environment where user latency is critical and rich high fidelity immersive audio environment is the goal.

Since the bit allocation is set so that the quantization for all sub-bands is the same, McDowell need not dequantize extracted sub-band data. It could be argued perhaps that McDowell

still dequantizes the data (just to a standard value), but McDowell certainly does not disclose such dequantization, and since the quantization of all of the sub-bands is the same, McDowell need not do so.

Claim 1 also recites that the extracted sub-band data is denormalized. This step is clearly not performed in the McDowell reference. As described in the above passage, McDowell teaches that instead of doing anything with the sub-band data, they simply look at the scale factor values that relate to the sub-band data. Larger scale factor values mean that the sub-band data is larger, and smaller scale factor values mean that the sub-band data is smaller.

> More specifically, the rendering process commences by unpacking and decompressing each component's scale factors into memory on a frame-by-frame basis (step 56), or alternately multiple frames at a time (see FIG. 7). At this stage only the scale factor information for each subband is required to assess if that component, or portions of the component, will be audible in the rendered stream. Since fixed length coding is used, it is possible to unpack and decompress only that part of the frame that contains the scale factors thereby reducing processor utilization. For SIMD performance reasons each 7-bit scale factor value is stored as a byte in memory space, and aligned to a 32-byte address boundary to ensure that a cache line read will obtain all scale factors in one cache fill operation and not cause cache memory pollution. To further speed this operation, the scale factors may be stored as bytes in the source material and organized to occur in memory on 32 byte address boundaries.

This allows McDowell to do the processing quicker than is the case if the sub-band data is denormalized using the scale factor, then examined to find the audio level. Therefore, McDowell does not teach denormalizing the extracted sub-band data and measuring an audio level for the extracted sub-band data. Nor does it teach measuring an audio level for the sub-band data. It teaches doing so from the scale factors.

The problem with the McDowell technique is that it can be used to determine a rough estimate of loudness for psychoacoustic purposes (and that is what they use it for), but it is not an accurate indication of the audio level for the extracted sub-band data.

Claims 10 and 19 recite analogous features and are patentable for the same reasons.

Claims 8, 17, and 26 recite the features of claims 1, 10 and 19 and are patentable for the same reasons.

In paragraph (7), the Office Action rejects claims 2, 11, and 20 under 35 U.S.C. § 103(a) as unpatentable in view of McDowell, Friedman, Petrillo, and Fiocca

With Respect to Claims 2, 11 and 20: Claim 2 recites:

The method of claim 19, further comprising using a psychoacoustic model in determining a perceived level of the measured audio signal according to human ear sensitivity.

The Office Action notes that McDowell discloses using psychoacoustic measurements to determine perceptually irrelevant information according to human sensitivity, and argues that Fiocca discloses using a psychoacoustic model to determine a perceived audio signal level to cut out unnecessary data in an audio signal and thereby reduce the computational load in the processor.

However, both McDowell and Fiocca teach applying the psychoacoustic model on a sub-band by sub-band basis, and before measuring the measurement of the audio level. This teaches away from claim 1 which recites that the psychoacoustic model is applied to the measured audio signal, not to individual sub-bands.

The proffered reason to modify McDowell (to cut out unnecessary data in an audio signal, thereby reducing the computational load on the processor) is illusory. Both McDowell and Fiocca teach applying the psychoacoustic model on a sub-band by sub-band basis for the very reason that this technique reduces the computational load on the processor. Specifically, McDowell recites:

> Second, DTS Interactive unpacks the scale factors (step **120**) and uses them in a simplified psychoacoustic analysis (see FIG. **9**) to determine which of the audio components selected by the map function (step **54**) are audible in each subband (step **124**). A standard psychoacoustic analysis that takes into account neighboring subbands could be implemented to achieve marginally better performance but would sacrifice speed. Thereafter, the audio renderer unpacks and decompresses only those subbands that are audible (step **126**). The renderer mixes the subband data for each subband in the subband domain (step **128**), recompresses it and formats it for packing as shown in FIG. **4** (item **86**).
>
> The computational benefits of this process are realized from having to unpack, decompress, mix, recompress and pack only those subbands that are audible. Similarly, because the mixing process automatically discards all of the inaudible data, the gaming programmer is provided greater flexibility to create richer sound environments with a larger number of audio components without raising the quantization noise floor. These are very significant advantages in a real-time interactive environment where user latency is critical and rich high fidelity immersive audio environment is the goal.

In contrast, claim 1 recites a system wherein a perceived audio level is determined from the measured audio level, which was generated from the dequantized and denormalized extracted sub-band data.

Claims 11 and 20 recite analogous features and are patentable for the same reasons.

In paragraph 8, the Office Action rejects claims 4, 5, 13, 14, 22 and 23 as unpatentable in view of McDowell, Friedman, Petrillo and Kallergis. The Applicants respectfully traverse.

As the Office Action acknowledges, McDowell does not disclose measuring an audio level without reconstructing the audio signal using channel characteristics. The Office Action suggests that this feature is disclosed in Kallergis ... a reference directed to reduction of the overflying noise of propeller planes. Kallergis, however, teaches nothing regarding channel characteristics at all, and therefore cannot teach using channel characteristics to weight an instantaneous or an overall level, as recited in claims 4 and 5, respectively. Instead, Kallergis simply teaches that the sound pressure level of a propeller can be described in a weighted overall value.

> **No zero lines are plotted in the measurement diagrams in FIG. 1, or FIGS. 3 and 5, which are to be described below. The unweighted sound pressure level of the propeller is designated by L_P, the weighted level by L_{PA}. The weighted overall sound pressure level is L_A (FIG. 4).**

Kallergis is also non-analogous art, and the Applicants respectfully traverse the notion that one of ordinary skill in the art would refer to Kallergis when solving a problem similar to that of McDowell. The notion that it would be obvious to do so to give the overall more influence on the final output is conclusory ... the issue is why one of ordinary skill in the art would do so.

In paragraph 9, claims 7, 16 and 25 are rejected as unpatentable over McDowell, Friedman, Patrillo and Smith. Claims 7, 16 and 25 are patentable for the reasons described above with respect to the claims they depend upon.

In paragraph 6 (presumably, paragraph 10), claims 6, 15 and 14 are rejected as unpatentable in view of McDowell, Friedman, Patrillo, and Pai. Claims 6, 15 and 25 are patentable for the reasons described above with respect to the claims they depend upon.

VI. <u>Dependent Claims</u>

Dependent claims 2, 4-8, 11, 13-17, 20 and 22-26 incorporate the limitations of their related independent claims, and are therefore patentable on this basis. In addition, these claims recite novel

elements even more remote from the cited references. Accordingly, the Applicant respectfully requests that these claims be allowed as well.

VII. New Claims

New claims 28-33 are presented for the first time in this Amendment. For the reasons described above, new claims are patentable over the prior art of record. These claims also recite features that render them even more remote from the cited references. For example, with respect to claim 1, even if the sub-band data is dequantized (and the Applicants believe it is not), the audio level for all of the dequantized data is not measured. Instead, the audio level is not determined for all of the dequantized data ... only the data from sub-bands that are not thrown out.

VIII. Conclusion

In view of the above, it is submitted that this application is now in good order for allowance and such allowance is respectfully solicited. Should the Examiner believe minor matters remain that can be resolved in a telephone interview, the Examiner is urged to call Applicants' undersigned attorney.

The Director is authorized to charge Applicant's Deposit Account No. 50-0383 should any fees become due with this response.

Respectfully submitted,

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